

Design by R. Badenhausen

Please note: Readers' Circuits have not been tested or post-engineered by the Elektor Electronics design laboratory.

The author and designer of this circuit performed a series of tests on several 'standard' soundcards used in PCs. Although most of these were found to have good anti-aliasing filters, in quite a few cases the frequency linearity of the analogue inputs left much to be desired. In most cases, this deficiency is caused by a rather too small input capacitor, a digital input filter dimensioned for a too low frequency, or simply by the user himself unwittingly connecting a cheap input cable of about 3 metres length to the input, which then sees a far too large capacitance.

The circuit shown in **Figure 1** allows any negative deviations at both ends of the frequency spectrum to be ‘pulled straight’. In fact, it causes the frequency response to be lifted by a controlled amount at the two far ends. An example of this is shown in **Figure 2**.

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The circuit consists of two passive filters (each having two roll-off points) — one for amplifying the low frequencies and one for the high frequencies. The first filter consists of components R1/C1/R2/R3, the second, of R4/C2/R5. Behind each filter sits a buffer/amplifier (IC1a and IC2a respectively) which serves to compensate the attenuation introduced by the filter. The Right (R) channel is of identical design with the component reference numbers increased by 10.

The values of the resistors in the buffer stages follow these easy equations: $R_6 = R_2/R_3$; $R_7 = R_1$; $R_8 = R_5$; $R_9 = R_4$. The same goes for the Right channel: $R_{16} = R_{12}/R_{13}$, etc.

Two voltage regulators, a 78L06 and a 79L06 together with a number of reservoir and decoupling capacitors, ensure a stable supply voltage for the circuit.

Measurement procedure

Before we can have a look at the calculation of the filter component values, we need to measure the error (non-linear frequency response) introduced by the analogue input.

It is unusual for a manufacturer of a soundcard to supply exact measurement data covering the inputs and outputs of the product. So, we have to do that ourselves. The job requires an accurate sinewave generator with a frequency range of 10 Hz to 25 kHz and a sound editor program like Cool Edit. If necessary, an oscilloscope may be connected to measure the output level at different frequencies.

Connect the generator to the input of the soundcard under test, then adjust it for the highest output level at which the input is not overdriven (for example, 200 mV). Next, use the software to measure the output level at different frequencies. Using Cool Edit, that is done by selecting 'Record' and then checking the signal level indicated by the VU meters. The range of the meters may be adapted by right-clicking on them.

Dimensioning

The author employs an Excel spreadsheet to calculate the values of the

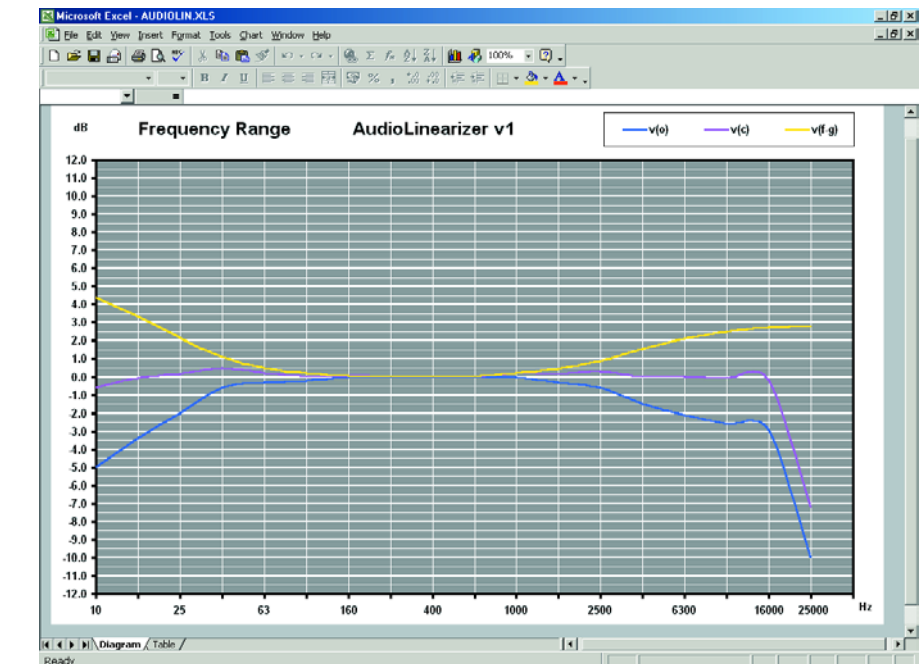


Figure 2. This graph shows the measured frequency response, the correction curve and the resulting curve.

components in the correction filters. The spreadsheet allows the measured values to be entered in column B (v(o), see Figure 3). The level at 1000 Hz is taken as the reference level (0 dB). Column A shows the frequencies at which the levels need to be measured. Column D shows

the response of the correction network of Figure 1, and column C, the resulting frequency response. Finally, in column E we see the attenuation v(f) per filter section — the respective values at 630 Hz and 1000 Hz indicate the attenuation which is to be compensated by the associated buffer/amplifier.

At the right-hand side of the Excel spread-

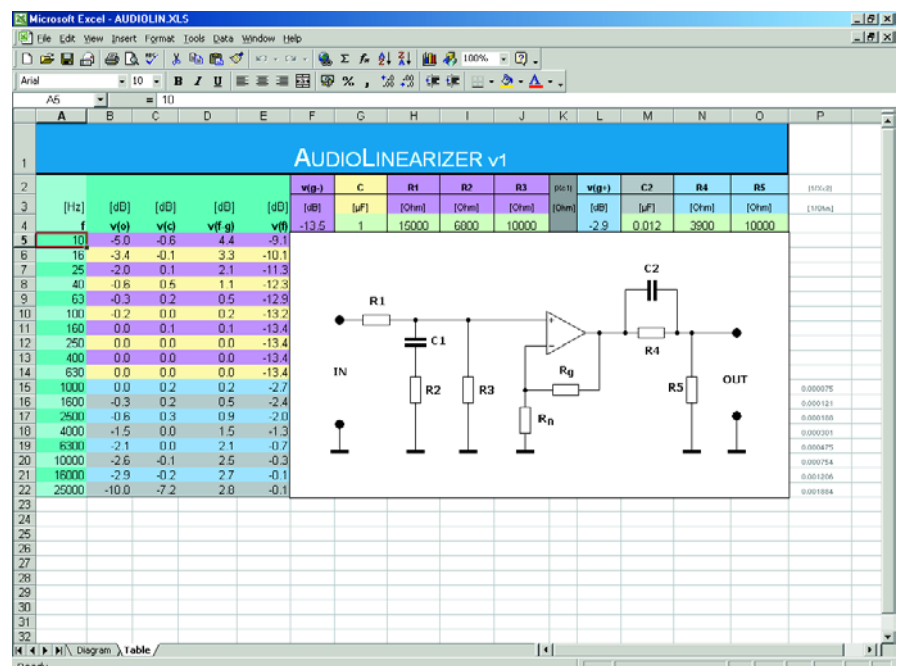


Figure 3. This Excel spreadsheet allows measured values to be entered and filter component values to be adapted to requirements. The associated curves (Figure 2) show the results instantly.

sheet the frequency determining components are shown, with an opamp thrown in for the necessary gain. The values of the filter components may be entered above this section, whereupon the spreadsheet will instantly compute the associated correction curve.

If you click on the 'Diagram' tab at the bottom, the three frequency response curves will appear: (1) the characteristic of the filter with the required values; (2) the measured response of the input and (3) the result of combining these two. Now, you are in a position to experiment with the filter components until the resultant frequency response is as flat as possible.

When the spreadsheet is started, you'll find that a number of component values and an example curve are already present. These may be edited as required.

Some rules of thumb are in order at this point to help you get the dimensioning right. A minimum value of 15 k Ω should be observed for R1, to prevent the output of the sound source affecting the filter. The ratio of R1 to R2//R3 determines the amplification of the low frequencies. A higher value of R1 causes a bass boost. The value of C1 determines the frequency at which the correction starts to act. Lowering the value of C1 causes the roll-off frequency to be increased. In the High section the ration of R4 to R5 determines the amplification of the higher frequencies, while C2 provides the onset of the high roll-off point. Here, a larger value results in a lower roll-off point.

Of course, all components in the filter sections will interact to some extent. Fortunately, you'll soon see the effects and results when actually entering values into the spreadsheet.

If only one correction is required, (i.e., for high or low only) the other section and the associated buffer/amplifier may be omitted.

The Excel spreadsheet is available from this month's Free Downloads page on our website, look for file number **020184-11**. Readers without access to the Internet may order it on floppy disk using the same number.

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